REPORT DOCUMENTATION PAGE

Public reporting burden for this collection of information is estimated to average 1 hour per response, including the time for reviewing instruc the data needed, and completing and reviewing this collection of information. Send comments regarding this burden estimate or any other a reducing this burden to Washington Headquarters Services, Directorate for Information Operations and Reports, 1215 Jefferson Davis Highv. Management and Budget, Paperwork Reduction Project (0704-0188), Washington, DC 20503

1. AGENCY USE ONLY (Leave blank)	2. REPORT DATE 16 June 2004	3. REPORT TYPE AND DATES COVERED Final Technical Report 1 Apr. 03 – 31 Mar 04

4. TITLE AND SUBTITLE INSTRUMENTATION FOR BASIC RESEARCH IN COMMUNICATION AND HEARING PROTECTION SYSTEMS	5. FUNDING NUMBERS DURIP AWARD TOPIC 01-012 F49620-03-1-0248

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9. SPONSORING / MONITORING AGENCY NAME(S) AND ADDRESS(ES) AIR FORCE OFFICE OF SCIENTIFIC RESEARCH, 4015 WILSON BLVD ARLINGTON, VA 22203-1954	10. SPONSORING / MONITORING AGENCY REPORT NUMBER

11. SUPPLEMENTARY NOTES

12a. DISTRIBUTION / AVAILABILITY STATEMENT APPROVED FOR PUBLIC RELEASE DISTRIBUTIION UNLIMITED	12b. DISTRIBUTION CODE A

13. ABSTRACT *(Maximum 200 Words)*
This DURIP grant supported instrumentation to perform basic research in control theory, hearing protector design, and psychoacoustics. The research is aimed at developing and evaluating hearing protection and communication systems for environments in which the user is exposed to high levels of nonstationary noise. The instrumentation provided by this grant consists of an Artificial Head Measurement Systems, a Low Frequency Acoustic Test Cell, a vibration isolation table, and digital signal processing equipment for rapid prototyping of active noise reduction (ANR) algorithms. During the performance period, the instrumentation directly supported graduate and undergraduate research work on hybrid ANR system and speech intelligibility metrics. Tests of hybrid ANR systems, using both feed-forward and feed-back control, were conducted using aircraft noise.

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20040709 035

14. SUBJECT TERMS Hearing Protection, Noise, Active Noise Reduction	15. NUMBER OF PAGES 12
	16. PRICE CODE

17. SECURITY CLASSIFICATION OF REPORT Unclassified	18. SECURITY CLASSIFICATION OF THIS PAGE Unclassified	19. SECURITY CLASSIFICATION OF ABSTRACT Unclassified	20. LIMITATION OF ABSTRACT

NSN 7540-01-280-5500

Standard Form 298 (Rev. 2-89)
Prescribed by ANSI Std. Z39-18
298-102

FINAL REPORT

INSTRUMENTATION FOR BASIC RESEARCH IN COMMUNICATION AND HEARING PROTECTION SYSTEMS

Period: April 1, 2003 – March 31, 2004

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Abstract

This DURIP grant supported instrumentation for performing basic research in control theory, hearing protector design, and psychoacoustics. The research is aimed at developing and evaluating hearing protection and communication systems for nonstationary, high noise environments and understanding human factors associated with advanced hearing protection.

1. Acquired Instrumentation and Costs

The instrumentation purchased consists of an Artificial Head Measurement System, an in-house designed and constructed Low Frequency Acoustic Test Cell (LFATC), a vibration isolation table, and digital signal processing (DSP) instrumentation for rapid prototyping of active noise reduction algorithms based on feedforward adaptive filters. The Artificial Head provides transfer function characteristics corresponding to that of the human ear in order to evaluate hearing protection systems under development objectively. It has signal playback ability to correlate objective noise reduction measurements within the Artificial Head environment with subjective auditory evaluation of human subjects. The DSP instrumentation allows rapid optimization of hearing protection hardware and active control algorithms in an environment that simulates the human auditory system as closely as possible, yet is more controllable and less expensive than direct human subject testing. The LFATC augments an existing test cell to provide binaural capabilities in experimental evaluation of hearing protectors. The vibration isolation table provides the lowest possible noise floor for the test cell environment.
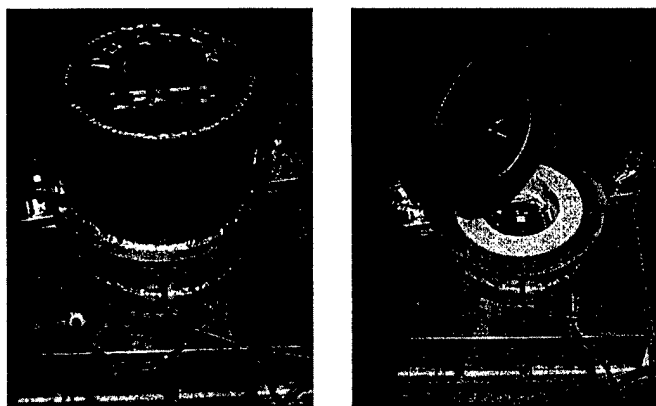

Figure 1 Second Generation Low Frequency Acoustic Test Cell for flat-plate evaluation of communication and hearing protection devices
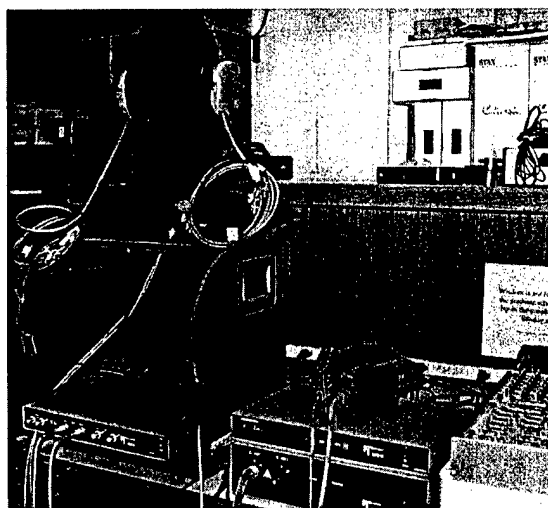

Figure 2 HMS II.3 Artificial Head Measuring System, Binaural playback system, and DSP development system

Proposed and actual instrumentation purchased, including costs and vendors are listed in Table 1. Figure 1 shows the second generation LFATC designed and constructed, and Figure 2 shows the HEAD Measuring system and a portion of the DSP instrumentation.

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Table 1 Original Cost Proposal and Actual Expenditures

Proposed Item/description	Quantity	Unit price	Subtotal	Total cost
HEAD Acoustics Artificial Head Measuring System				$44,715
HMS II.3 HEAD measurement system w/ right ear and mouth	1	$16,675	$16,675	
HIS L Impedance simulator, left for HMS II.3	1	$4,190	$4,190	
BEQ I Digital electronics for HMS II.3	1	$10,725	$10,725	
PEQ IV High voltage power amplifier	1	$3,328	$3,328	
HPS IV Playback System	1	$6,305	$6,305	
HA III Electrostatic Playback headphone for use with HPS IV	2	$790	$1,580	
ECS 1.0 Artificial head ear inserts	1	$1,060	$1,060	
VX PCMCIA Stero sound card for playback system	1	$852	$852	
Binaural LFATC				$17,390
Materials	1	$3,000	$3,000	
Labor	1	$4,500	$4,500	
Instrumentation				
B&K 4192 Precision microphone and 2669 pre-amplifier	4	$1,600	$6,400	
Noise source speaker & power amp (B&K 2716)	2	$950	$1,900	
Furman Q602-B 30-Band Acoustic equalizer	2	$795	$1,590	
Vibration isolation table	1	$4,140	$4,140	$4,140
dSPACE DSP System and Host Computer				$17,820
DSP board, software, and connector panel	1	$15,570	$15,570	
Host computer	1	$2,250	$2,250	
Total				$84,065

Actual Item Purchased	Quantity	Unit price	Subtotal	Actual cost	Vendor
HEAD Acoustics Artificial Head Measuring System				$45,909	
HMS II.3 HEAD measurement system w/ right ear and mouth	1				HEAD Acoustics
HIS L Impedance simulator, left for HMS II.3	1				HEAD Acoustics
BEQ I Digital electronics for HMS II.3	1				HEAD Acoustics
PEQ IV High voltage power amplifier	1				HEAD Acoustics
HPS IV Playback System	1				HEAD Acoustics
HA III Electrostatic Playback headphone for use with HPS IV	2				HEAD Acoustics
ECS 1.0 Artificial head ear inserts	1				HEAD Acoustics
Flexible Pinnae for HMS II.3	2				HEAD Acoustics
VX PCMCIA Stero sound card for playback system	1		$493		BSW-USA
Binaural LFATC				$15,042	
Materials			$1,743		
Labor			$1,941		
Instrumentation					
B&K 4192 Precision microphone and 2669 pre-amplifier	4		$9,443		B&K Crutchfield,
Noise source speaker & power amp	2		$1,630		DeMatteo Music
30-Band Acoustic equalizer (multi-channel)	1		$285		Local vendor
Vibration isolation table	1			$2,458	Kinetic Systems
dSPACE DSP System and Host Computer				$20,973	dSPACE, Dell, Mathworks
Total				$84,381	

2. Contributions of Instrumentation to Hearing Protection and Communication Research

During the performance period, the instrumentation directly supported the work of M.S. candidate Alex Streeter, who developed a hybrid ANR system, B.E. candidates Katherine Baus, who evaluated speech intelligibility metrics for hearing protectors, Kenneth Leon, and Faris Raman, who investigated the role of cancellation speaker dynamics on low frequency ANR performance, and M.S. candidate Matthew Maher, who is supported though an AFOSR Phase II STTR grant with Creare, Inc. A brief review of research activities is provided here.

2.1 Feedforward Adaptive Control for Hearing Protection

Least mean square (LMS) feedforward ANR algorithms have stability and performance issues related to insufficient excitation, nonstationary noise fields, finite-precision arithmetic, quantization and measurement noise, low signal-to- noise ratio, and uncertain plant dynamics. These uncertainties - in both the system dynamics and signals - often cause instability in the conventional LMS filter. The leaky LMS filter addresses stability by introducing a leakage parameter to leak excess energy in the LMS weight update equation associated with these factors. However, a constant, manually selected tuning parameter does not suffice for noise sources ranging in degree of stationarity and over large acoustic dynamic ranges evident in real-world applications of noise cancellation systems. Hence, currently, "worst case" noise environment scenarios must be used to empirically select tuning parameters, resulting in substantial performance loss over a range of possible noise field characteristics.

The Lyapunov design method developed by the investigators in [1,2] enhances both stability and performance of the leaky, normalized LMS algorithm. It accounts for nonpersistent excitation conditions and nonstationary reference inputs and requires no *a priori* knowledge of the reference input signal characteristics other than a lower bound on its magnitude or a minimum signal-to-noise ratio. Using the LFATC, DSP development system, and HEAD measuring system, M.S. candidate Alex Streeter developed and evaluated a hybrid ANR implementation comprised of a Lyapunov-tuned LMS filter and a digital feedback ANR system. The hybrid system and performance results are detailed in [3]. The overall system, shown in Figure 3, aims to further improve noise reduction performance for nonstationary noise sources and to increase stability margins. A digital feedback system is designed to provide low level, broadband performance, independent of the noise source. The feedforward system acts on the resulting error signal to further increase noise attenuation. Unlike previous studies, where feedforward ANR is hybridized with a commercial narrowband analog controller, we develop a broadband feedback controller digitally. The presence of this feedback system is shown to increase feedforward gain stability margin substantially and to reduce sensitivity of overall performance on the temporal characteristics of the noise source.
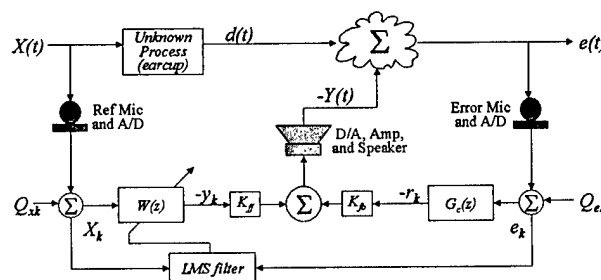

FIG. 3. Combined feedforward-feedback topology

Experimental evaluation of the hybrid feedforward-feedback system was conducted in the Low Frequency Acoustic Test Cell (LFATC). The test cell acts as a one-dimensional waveguide and is designed to have a flat (to ±1 dB) acoustic frequency response from 10 to 200 Hz. Digital

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equalization extends this range to approximately 1600 Hz. A single earcup is mounted over the base plate of the test cell with an airtight seal. The test cell instrumentation includes: (1) a 15.2 cm diameter 100 W speaker mounted in the top plate of the cell to provide the noise signal (up to 140 dB); and (2) two precision Brüel & Kjær 4190 Type I microphones. One microphone is mounted through the sidewall of the test cell for source level measurement and the other is mounted axially in the base plate under the earcup to represent the location of the external opening to the ear canal. Noise floors of these precision microphones average 53 dB and 48 dB, respectively, in the measurement range 40-1250 Hz

The test device consists of an earcup taken from a commercial feedback ANR headset. Existing hardware within the earcup includes a noise cancellation speaker, an electret error microphone, a communication speaker (not used in this study), and feedback ANR circuitry (also not used in this study). Without disturbing the damping materials that provide passive noise attenuation, a 0.500" hole was drilled in the shell to add an external reference electret microphone. The two microphones are conditioned through preamplifiers developed in-house, which provide a noise floor of 50 dB in the measurement range 40-1250 Hz and a dynamic range of at least 75 dB. When mounted on the base of the test cell, the earcup's error and reference microphones are calibrated with respect to the precision Brüel & Kjær microphones mounted in the base and side of the test cell.

Four noise sources were selected for the performance evaluation: (1) Individual pure tones at 1/3-octave center frequencies from 40 Hz through 1250 Hz, (2) a sum-of-tones signal comprised of 1/3-octave pure tones between 50 Hz and 800 Hz, (3) F-16 cockpit noise band-limited between 50 Hz and 800 Hz, and (4) Huey helicopter noise likewise band-limited between 50 Hz and 800 Hz. These noise sources can be viewed as increasingly less ideal operating conditions. Pure tones are the most ideal operating condition as they allow the cancellation gains K_{ff} and K_{fb} in Figure 3 to be optimally tuned for each frequency, whereas for all other noise sources one value for each gain is applied to all frequencies. The two minute noise source recording for F-16 noise used in experiments exhibits significant temporal variation. Huey helicopter noise resembles F-16 noise, but with the addition of both tonal components (a 55 Hz fundamental and associated harmonics) and impulsive staccato components in the time domain from the rotor blades passing 10.7 times per second. The first three noise sources are set to an average level of 110 dB, whereas the fourth noise source (Huey helicopter) is set to 105 dB to avoid distortion in the cancellation speaker. All noise levels are reported in dB relative to 20 μPa, with no weighting applied. Passive, active and total ANR performance are measured. The hybrid controller is implemented the dSPACE DS1103 controller board at an update frequency of 10 kHz; the LMS filter length is 500 taps. All reduction performance data are given as the insertion loss between the precision microphone outside of the earcup and the one inside in the base of the test cell. Thus, they account for the separation path between the noise source and the wearer's ear.

Figure 4 shows the active attenuation in dB for individual tones, as measured by the B&K precision microphone located inside the earcup. The results show that the feedback system has low level (5-10 dB) but high bandwidth noise reduction capabilities. In contrast, the feedforward system performs exceptionally well in the range 80-400 Hz, with diminished performance above and below that range. Whereas both systems have only moderate to good attenuation at low

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(<100 Hz) frequencies, the combined system is able to provide approximately 30 dB of active attenuation at these frequencies. *Combining the two independent systems has resulted in performance that is greater than the sum of its parts.* Additionally, whereas both the feedforward and feedback systems add noise above 700 Hz, the combined system provides positive attenuation throughout the frequency range 40-1250 Hz. The resulting noise level inside the earcup has been reduced by a total of 36 to 51 dB within the 40-1250 Hz band. Total noise reduction, which include passive attenuation, causes the error microphone signal to approach its noise floor, thus the noise reduction performance approaches its physical limits.

FIG. 4. Active Attenuation performance of each ANR system in response to puretone noise.

Figure 5 shows the active attenuation of the sum-of-tones noise source. In this and subsequent cases, the feedforward and feedback gains can assume only one value for all frequencies, whereas for individual tones, K_{ff} and K_{fb} can be tuned to an optimal value for each frequency. The results show that the individual tones are successfully attenuated by as much as 28 dB by the hybrid system. The hybrid system exhibits the same synergistic performance improvement over the independent feedforward and feedback systems. Whereas the feedback system provides an average of only 7.8 dB of active attenuation, and the feedforward an average of 16.6 dB, the hybrid system provides an average of 27.2 dB of active attenuation. When combined with the earcup's passive attenuation, this means that an average source level of 110 dB is successfully reduced to 70.6 dB, a level that is considered safe for long periods of exposure.

FIG. 5. Active Attenuation performance of the ANR systems when subjected to Sum-of-Tones noise

Figure 6 presents the active attenuation for each system when subjected to F-16 aircraft noise. This noise source most closely resembles band-limited white noise in that it contains minimal tonal content and, over long time periods, has a fairly uniform spectral component. However, during short periods of time its spectral content shifts considerably, which presents problems for traditional LMS filters. Despite the difficulties associated with this nonstationary noise source, the results show that the hybrid system provides an average active attenuation of 17.3 dB (32 dB total attenuation), reducing the 110 dB source level to 78.2 dB. Once again, the results show that
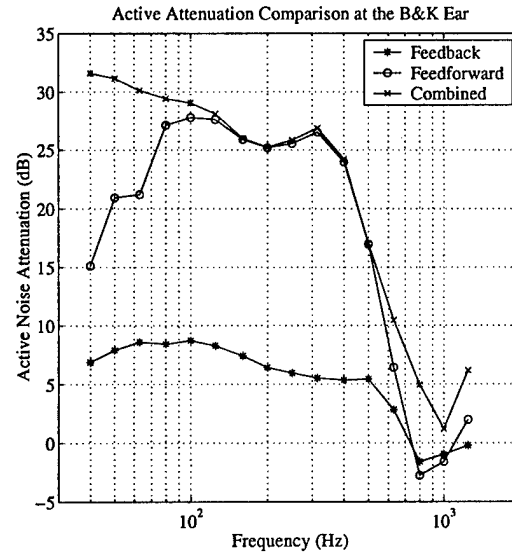
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the hybrid system has substantially greater performance than either of the independent systems acting alone, particularly for frequencies less than 200 Hz. Additionally, whereas the feedforward system *added* noise for frequencies above 500 Hz, the hybrid system largely avoided adding any noise in the 50-800 Hz band.

Lastly, the systems were subjected to Huey helicopter noise. This noise source contains broadband nonstationary components like the F-16 noise, but also has a tonal component following a 55 Hz fundamental attributed to the tail rotor, and a temporal component due to blade passage. This temporal component is a periodic broadband impulse, rather than a low-frequency harmonic. In order to keep this periodic impulse from forcing the ANR systems to over-drive the cancellation speaker, the source level is reduced to 105 dB. The active attenuation results are shown in Figure 7. Once again, the addition of the feedback system to the feedforward system significantly improved the low-frequency attenuation, in this case by 5-10 dB. The tonal component is eliminated by both the feedforward and hybrid system, but largely untouched by the feedback system. The feedback system was unsuccessful in removing the temporal component of the helicopter noise; the feedforward system could not completely remove it, either. In contrast, the combined hybrid system is able to almost completely remove the periodic noise leaving behind a broadband background noise whose average level was 77.4 dB. Table 2 summarizes these performance results, showing average source, passive, active, and total noise reduction performance for each noise source.
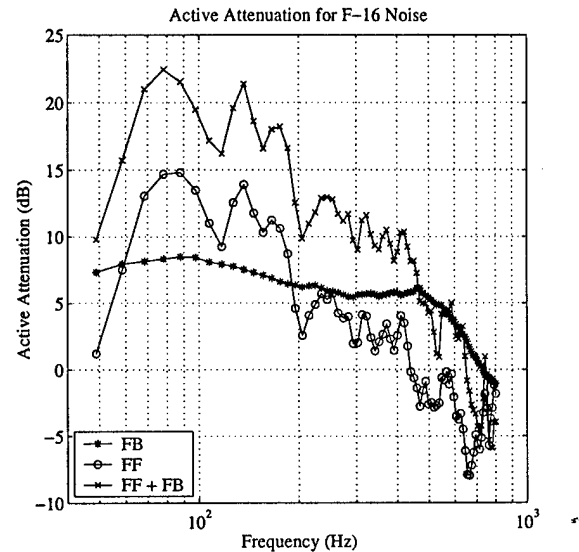

FIG. 6. Active Attenuation performance of the ANR systems due to F-16 cockpit noise
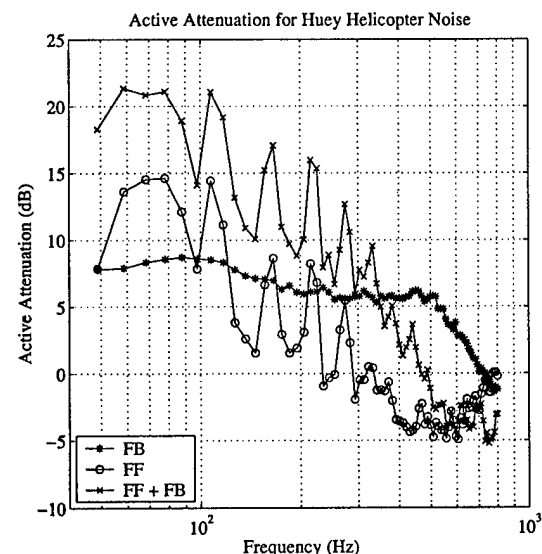

FIG. 7. Active Attenuation performance of each ANR system due to Huey Helicopter noise

Table 2. Summary of passive, active, and total noise reduction performance for feedback, feedforward, and hybrid ANR

Noise Source	Average Noise Level (dB)				Total Attenuation (dB)			Active Attenuation (dB)		
	Source	Passive	Feedb.	Feedf.	Hybrid	Feedb.	Feedf.	Hybrid	Feedb.	Feedf.	Hybrid
Sum-of-tones	110.3	97.8	90.0	81.2	70.6	20.3	29.1	39.7	7.8	16.6	27.2
F-16 Cockpit	110.3	95.4	87.6	85.6	78.2	22.7	24.6	32.1	7.9	9.8	17.3
Huey Cockpit	105.3	94.2	86.0	83.7	75.8	19.3	21.6	29.5	8.2	10.5	18.4

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The addition of feedback to the feedforward system in the hybrid system not only improves active performance, but it also improves the gain margin of the individual systems. In the feedback system, increasing the path gain, K_{fb}, generally increases the feedback attenuation. However, the gain that provides maximum noise attenuation is extremely close to the threshold of instability, forcing a stability-performance tradeoff common in commercial feedback systems. In a similar way there is a maximum feedforward gain, K_{ff}, above which the weight vector $W(z)$ grows without bound, or overexcites certain frequencies (particularly 700-800 Hz). When the two systems are combined, both gains can be increased to levels that otherwise would cause instability. When this happens, the increased gain allows for higher overall active attenuation. Stated

FIG. 8. Maximum stable gains (K_{ff}) of the feedforward system

differently, the feedforward and feedback systems in the hybrid system can provide the same attenuation levels at the same gain values as before, but now have larger stability margins. For the feedback system, adding the feedforward system allows K_{fb} to be increased by approximately 20% before instability reoccurs. However, the increased stability is most notable in the feedforward system. Figure 8 shows the *maximum stable* (not necessarily optimal) feedforward gain K_{ff}, as determined experimentally, as a function of frequency. As Fig. 8 shows, augmenting the feedforward system with feedback allows the maximum stable K_{ff} to be increased by, at some frequencies, orders of magnitude.
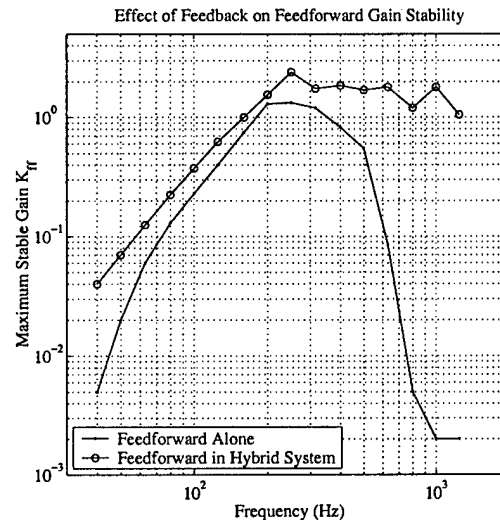
2.2 Effect of Speaker Dynamics on ANR

In prototype headsets used in development of feedforward ANR methods, off-the-shelf speaker and microphones were used, and an off-the-shelf earmuff shell and seal housed these components. The best available speaker identified for low frequency noise cancellation had an experimentally determined transfer function with a roll off of approximately 50 Hz and speaker distortion below 50 Hz. B.E. candidates Kenneth Leon and Faris Raman investigated the effect of speaker dynamics on ANR performance. Their work characterized a variety of off-the-shelf speakers and evaluated the use of a filtered-X LMS filter for "flattening" the transfer function from the speaker to error microphone, versus using digital equalization to flatten the response. The filtered-X LMS algorithm is required when the dynamics of the speaker transfer function vary, e.g., due to manufacturing differences, temperature, humidity, or packaging variations. This algorithm adaptively models the speaker transfer function so as to account for the frequency dependent cancellation path gain. A digital equalizer, which consists of a fixed infinite-impulse-response transfer function can also be used to flatten the speaker response in cases where differences between speaker performance in a sample of manufactured earcups is modest. This option is less intensive computationally, but it introduces phase shift in the overall speaker-equalizer response. Kenneth and Faris implemented both methods and tested each within the feedforward ANR system and found that a simple digital filter was as effective as a filtered-X LMS filter at providing a flat cancellation path response and incurred substantially less computation.

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2.3 Speech Intelligibility of Hearing Protection Devices

As part of a NIOSH Hearing Impaired Worker Project, research at Ohio State University was directed towards development of a high sensitivity clinical evaluation protocol for evaluating speech intelligibility in noise for conventional and uniformly attenuating passive hearing protectors [4]. Two commercially available sentence-based clinical tests of speech intelligibility in noise – the Quick Speech in Noise (QSIN) [5] and the Hearing in Noise Test (HINT) [6] – considered to have a sufficient number of test items for the study were evaluated along with the Diagnostic Rhyme Test (DRT) [7]. The QSIN test presents a list of six sentences with five key words per sentence in four-talker babble noise. Sentences are presented at pre-recorded SNRs of 25, 20, 15, 10, 5, and 0 dB, providing the ability to measure speech intelligibility in noise for subjects with normal hearing as well as those with severe hearing impairments. The DRT employs a two alternative forced choice paradigm and provides more than four times the number of items than the traditional modified rhyme test. A test protocol was developed to choose the most sensitive indicator of speech intelligibility in noise from among the three methods. Results of testing with eight normal hearing subjects indicated that QSIN was the most sensitive test, followed by the HINT and DRT tests, for each listening condition evaluated.

This work formed a basis from which to select metrics and to begin to develop test protocols for evaluating speech intelligibility afforded by feedforward ANR HPDs. Katherine Baus acquired, modified, and evaluated the QSIN test for measurement of speech intelligibility of ANR hearing protectors as part of the Bachelor of Engineering capstone project. The QSIN test, with its four talker babble background noise and ample test sentences provides a sensitive measure of speech intelligibility; however, since the background noise is in the speech bandwidth, it also can mask the effect of ANR on speech intelligibility. The ability to produce high levels of low frequency noise reduction through ANR should have a significant impact on the upward spread of masking. It is known that low frequency masking noise influences speech intelligibility by creation of distortion products within the cochlea, and that the relationship between noise intensity and the growth of masking is nonlinear [8].

Using the QSIN test sentences, Katy configured tests with two different background noises – four talker babble and pink noise, designed experiments to compare speech intelligibility in a passive and active hearing protector, and administered the test to ten subjects. While her subject sample (and levels of noise exposure) was insufficient to draw rigorous conclusions, this initial testing provided evidence that ANR improves speech intelligibility through reduction of low frequency noise and will form the basis for future studies.

2.4 Development of Hearing Protection for Extreme Noise Fields

ANR in very high noise fields requires new approaches to modeling and design of hearing protection. In high noise fields, sound enters the cochlea through the air transmission pathway, cerebral-spinal fluid, conduction through the bones of the skull, and secondary bone conduction through other body paths. In addition to passive and active noise reduction of the air transmission path, active structural control and passive noise control through design of the helmet structure may be required to reduce bone conducted noise through the skull. Current AFOSR sponsored research focuses on developing measurement methods and models for quantifying the contributions of air transmitted and bone conducted sound to the inner ear [9]. In support of this Phase II STTR research between Creare, Inc., and Dartmouth College, the instrumentation from this DURIP award has been used extensively to evaluate the performance of many helmets and

headsets for maximizing noise attenuation of the air conduction path, and for developing models of the structrural/vibrational characteristics of the human skull. Extensive impedance response data have been collected from a human skull simulator constructed at Creare. Three "black box" modeling approaches have

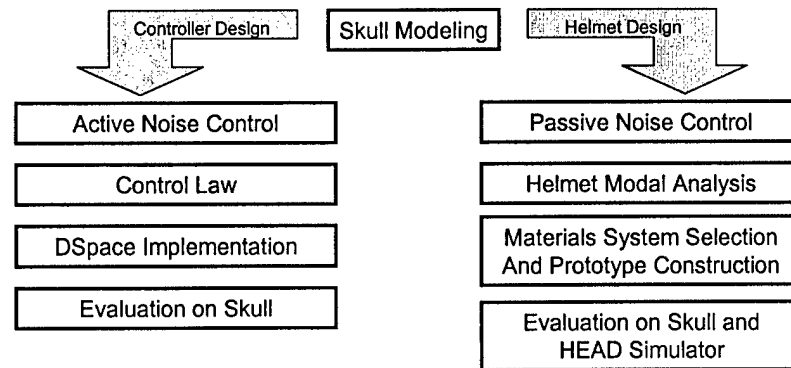|		
Controller Design	Skull Modeling	Helmet Design
Active Noise Control		Passive Noise Control
Control Law		Helmet Modal Analysis
DSpace Implementation		Materials System Selection And Prototype Construction
Evaluation on Skull		Evaluation on Skull and HEAD Simulator

Figure 7 Noise reduction approaches for extreme noise environments

been developed to provide a transfer function between a bone oscillator exciting the skull and accelerometer output at various positions on the skull. Based on these skull vibration models, which show a large first mode between approximately 800 and 2000 Hz, two design paths to attenuating bone conducted sound are proposed, as summarized in Figure 7, each beginning with models of skull vibration that have been developed. The first path is to design an active vibration suppression system that reduces the first resonance peak of the skull, using a virtual passive controller, or a notch filter. These approaches are being implemented using the dSPACE instrumentation. Passive control seeks to design a helmet structure that, when mounted on the human head, attenuates the first mode to the extent possible by minimizing helmet vibration in that mode. In the final months of the project, a prototype helmet is to be constructed. Its vibration characteristics will be measured using Creare's human skull simulator, and its passive attenuation characteristics will be evaluated using the HEAD measuring system. Matt Maher is designing both the active system and the passive helmet structure as part of his M.S. and Master of Engineering Management degree programs.

3. Summary
Instrumentation for basic research in communications and hearing protection systems has been implemented, and a number of undergraduate and graduate research projects have been supported through this instrumentation over the past year. It is expected that in coming years, we will use the instrumentation to continue development of hearing protection devices, including in-ear and supra-aural devices, continue to develop and evaluate psychoacoustic and speech intelligibility metrics, and move into development of next generation digital signal processing methods for hearing protection and hearing augmentation.

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